

AMENDMENTS TO THE CLAIMS

Claims 1, 3, 4, 9- 20, 22-25, 27-34, 36, 38-52, 55, 57-67, and 70-75 remain as previously pending. Please cancel Claims 5 and 7. Please amend Claim 8.

1. (Previously Presented) A system for enhancing intelligibility of a voice signal that is degraded by factors that reduce intelligibility of the voice signal, said system comprising:

an input configured to receive a voice signal that includes human spoken words;

an aural filter operatively coupled to said input, said aural filter configured to filter said voice signal to produce a filter output signal wherein low frequencies below speech frequencies and high frequencies above speech frequencies are attenuated with respect to speech frequencies;

a speech expander operatively coupled to said aural filter to produce an expanded signal, said speech expander configured to amplify said filter output signal according to an amplifier gain, wherein said amplifier gain is a function of an envelope amplitude of said filter output signal; and

a combiner configured to combine at least a portion of said expanded signal and at least a portion of said voice signal to produce an enhanced signal representing said spoken words;

wherein, when the voice signal is operating a high volume levels, the system emphasizes middle speech frequencies over low and high frequencies; and

wherein, when the voice signal is operating at low volume levels, the system provides more low and high frequency components of the voice signal than when the voice signal is operating a high volume levels;

such that the system provides a transfer function which approximates an inverse of the transfer function of human hearing.

2. (Canceled)

3. (Original) The system of Claim 1, wherein said speech expander comprises an envelope detector and a gain controlled amplifier, wherein at least a

portion of said filter output signal is provided to an input of said envelope detector configured to detect an envelope amplitude of said at least a portion of said filter output signal.

4. (Original) The system of Claim 1, wherein said amplifier gain increases according to an attack time constant and said amplifier gain decreases according to a decay time constant.

5. (Canceled)

6. (Canceled)

7. (Canceled)

8. (Currently Amended) ~~The communication device of Claim 5, A~~
communication device for sending voice information to a communication receiver,
where the voice information may become contaminated by noise that reduces the
intelligibility of the voice information, said communication device comprising:

a sender configured to send a voice signal comprising words spoken by a
person over a communication channel; and

a voice enhancer operably connected to said sender, said voice enhancer
comprising:

an aural filter operatively coupled to a voice signal in said sender,
said aural filter configured to filter said voice signal to produce a filter
output signal wherein low frequencies below speech frequencies and high
frequencies above speech frequencies are attenuated with respect to
speech frequencies;

a speech expander operatively coupled to said aural filter to
produce an expanded voice signal, said speech expander configured to
amplify said filter output signal according to an amplifier gain, wherein
said amplifier gain is a function of an envelope amplitude of said filter
output signal; and

a combiner configured to combine at least a portion of said
expanded voice signal and at least a portion of said voice signal to
produce an enhanced voice signal;

wherein said voice enhancer is configured to provide a transfer function that approximates an inverse of loudness contours for human hearing;

wherein said speech expander comprises a gain controlled amplifier; and

wherein the amplifier gain increases according to an attack time constant when said envelope amplitude has a positive slope and said amplifier gain decreases according to a decay time constant when said envelope amplitude has a negative slope.

9. (Previously Presented) A communication device configured to receive voice information from a communication sender, comprising:

a communication receiver configured to receive voice information comprising words spoken by a person from a communication channel; and

a voice enhancer operably connected to said communication receiver, said voice enhancer comprising:

an aural filter configured to filter an input signal to produce a filtered signal;

an expander comprising an amplifier configured to amplify said filtered signal to produce an amplified signal, wherein a gain of said amplifier is a function of an amplitude envelope of said filtered signal; and

a combiner configured to combine at least a portion of said amplified signal and at least a portion of said input signal to produce an output signal;

wherein said voice enhancer enhances formants of the voice information to increase intelligibility of the voice information; and

wherein said voice enhancer provides a transfer function that approximates a complement of Fletcher-Munson curves for tones in a frontal sound field for humans.

10. (Original) The communication device of Claim 9, wherein said communication device is a cordless telephone comprising a handset and a base unit.

11. (Original) The communication device of Claim 9, wherein said communication device is a cellular telephone.

12. (Original) The communication device of Claim 9, wherein said aural filter attenuates low and high frequencies with respect to middle frequencies.

13. (Previously Presented) The communication device of Claim 9, wherein said combiner adds at least a portion of said amplified signal to said input signal.

14. (Original) The communication device of Claim 9, further comprising a user control, said user control configured to enable and disable said voice enhancer.

15. (Original) The communication device of Claim 9, further comprising a user control, said user control configured to vary an amount of enhancement produced by said voice enhancer.

16. (Previously Presented) The communication device of Claim 9, wherein said voice enhancer is configured to approximate an inverse of loudness contours of human hearing.

17. (Previously Presented) An apparatus, comprising:

an aural filter configured to filter an input signal comprising words spoken by a person to produce a filtered signal;

an expander comprising an amplifier configured to amplify said filtered signal to produce an amplified signal, wherein a gain of said amplifier depends in part on an envelope of said filtered signal; and

a combiner configured to combine at least a portion of said amplified signal and at least a portion of said input signal to produce an output signal;

wherein said apparatus is configured to provide a transfer function that emphasizes middle speech frequencies over low and high frequencies at high volume levels and is flatter at low volume levels.

18. (Original) The apparatus of Claim 17, wherein said aural filter attenuates low and high frequencies with respect to middle frequencies.

19. (Previously Presented) The apparatus of Claim 17, wherein said combiner adds at least a portion of said amplified signal to said input signal.

20. (Original) The apparatus of Claim 17, wherein a gain of said amplifier depends in part upon a property of said filtered signal.

Appl. No. : **09/185,876**
Filed : **November 3, 1998**

21. (Canceled)
22. (Original) The apparatus of Claim 17, wherein said aural filter attenuates low frequencies with respect to middle frequencies.
23. (Original) The apparatus of Claim 17, wherein a gain of said amplifier increases according to an attack time constant.
24. (Original) The apparatus of Claim 17, wherein a gain of said amplifier decreases according to a decay time constant.
25. (Original) The apparatus of Claim 17, wherein said aural filter attenuates low frequencies and high frequencies with respect to middle frequencies.
26. (Canceled)
27. (Original) The apparatus of Claim 17, operably connected to a recording device.
28. (Original) The apparatus of Claim 17, said apparatus incorporated into a telephone and adapted to improve intelligibility of voice information processed by said telephone.
29. (Original) The apparatus of Claim 17, said apparatus incorporated into a hearing aid and adapted to improve intelligibility of voice information processed by said hearing aid.
30. (Original) The apparatus of Claim 17, said apparatus incorporated into a public-address system and adapted to improve intelligibility of voice information processed by said public-address system.
31. (Original) The apparatus of Claim 17, said apparatus incorporated into a communication system and adapted to improve intelligibility of voice information processed by said communication system.
32. (Original) The apparatus of Claim 17, wherein said aural filter is an analog filter.
33. (Original) The apparatus of Claim 17, wherein said aural filter is a digital filter.
34. (Previously Presented) A method for enhancing intelligibility of voice information, comprising the steps of:

filtering at least a portion of a first signal that includes human voice sounds to produce a filtered signal having an amplitude envelope;

expanding at least a portion of said filtered signal using an amplifier having a variable gain to produce an enhanced signal;

detecting the amplitude envelope to produce a gain control signal to control the gain of the amplifier;; and

combining at least a portion of said first signal with said enhanced signal to produce an improved signal;

wherein the method emphasizes middle speech frequencies over low and high frequencies at high volume levels and is flatter at low volume levels, such that the method provides a transfer function which approximates an inverse of loudness contours for human hearing.

35. (Canceled)

36. (Currently Amended) The method of Claim ~~35~~34, wherein said step of combining comprises adding at least a portion of said first signal to said enhanced signal.

37. (Canceled)

38. (Previously Presented) The method of Claim 34, wherein said variable gain is a function of at least a portion of said filtered signal.

39. (Previously Presented) The method of Claim 34, wherein said variable gain is a function of at least a portion of an envelope of said filtered signal.

40. (Previously Presented) The method of Claim 34, wherein said variable gain is a function of at least a portion of an average power of said filtered signal.

41. (Previously Presented) The method of Claim 34, wherein said variable gain is a function of at least a portion of a square-root of the mean of the squares average of said filtered signal.

42. (Previously Presented) The method of Claim 34, wherein said variable gain depends upon at least a portion of an average peak value of said filtered signal.

43. (Previously Presented) The method of Claim 34, wherein said variable gain increases according to an attack time constant.

44. (Previously Presented) The method of Claim 43, wherein said variable gain decreases according to a decay time constant.

45. (Previously Presented) The method of Claim 44, wherein said attack time constant is shorter than said decay time constant.

46. (Previously Presented) The method of Claim 34, wherein said variable gain depends upon at least a portion of said first signal.

47. (Previously Presented) The method of Claim 34, wherein said step of filtering comprises filtering said first signal using an aural filter.

48. (Original) The method of Claim 47, wherein said aural filter comprises a bandpass filter.

49. (Original) The method of Claim 47, wherein said aural filter attenuates low frequencies and high frequencies with respect to middle frequencies.

50. (Original) The method of Claim 47, wherein said first signal comprises noise components and voice components, and wherein said aural filter combined with said speech expander reduces the degradation of said voice components by said noise components.

51. (Original) The method of Claim 34, further comprising the step of providing said enhanced signal to a loudspeaker system to be projected as sound into an area of ambient noise.

52. (Original) The method of Claim 34, further comprising the step of providing said enhanced signal to a recording device.

53. (Canceled)

54. (Canceled)

55. (Previously Presented) An apparatus for enhancing intelligibility of voice information, said apparatus comprising:

aural filter means for filtering an input signal to produce a filtered signal, said input signal containing human voice information;

gain controlled amplifier means for amplifying the filtered signal to produce an expanded signal;

gain control means for controlling a gain of the gain controlled amplifier as a function of an envelope amplitude of the filtered signal;

attack time means for increasing the gain for an attack time when a slope of the envelope amplitude is positive;

decay time means for decreasing the gain for a decay time when the slope of the envelope amplitude is negative; and

combiner means for combining at least a portion of said expanded signal with at least a portion of said input signal;

wherein said apparatus is configured to provide a transfer function that emphasizes middle speech frequencies over low and high frequencies at high volume levels and is flatter at low volume levels, such that said transfer function approximates an inverse of loudness contours for human hearing of tones in a sound field.

56. (Canceled)

57. (Previously Presented) An apparatus, comprising:

an input configured to receive an input signal comprising words spoken by a person; and

a dynamic filter configured to filter said input signal to produce an enhanced signal with modified voice components, said dynamic filter configured to provide a transfer function that depends at least in part on an envelope of the input signal, wherein said transfer function emphasizes middle speech frequencies over low and high frequencies at high volume levels and is flatter at low volume levels.

58. (Original) The apparatus of Claim 57, wherein said dynamic filter comprises a bandpass filter and an expander.

59. (Original) The apparatus of Claim 57, wherein said dynamic filter comprises an aural filter.

60. (Original) The apparatus of Claim 57, wherein said dynamic filter comprises a filter that attenuates low and high frequencies relative to middle frequencies.

61. (Original) The apparatus of Claim 57, wherein said dynamic filter comprises an expander.

62. (Previously Presented) The apparatus of Claim 57, further comprising a combiner configured to combine at least a portion of said input signal with at least a portion of said enhanced signal.

63. (Original) The apparatus of Claim 57, further comprising a user control, said control configured to allow a user to adjust a transfer function of said dynamic filter.

64. (Previously Presented) A method of improving the intelligibility of voice sounds contained within a signal source when the signal source is reproduced through a loudspeaker, said method comprising the following steps:

detecting an envelope of a signal source comprising words spoken by a person to produce a control signal;

filtering the signal source according to a frequency response related to human hearing characteristics to produce a filtered signal;

modifying the frequency response used to filter said signal source wherein the amount of modification is a function of the control signal; and

combining the signal source with the filtered signal to produce an output signal having enhanced voice sounds;

wherein, when the first signal is operating a high volume levels, the method emphasizes middle speech frequencies over low and high frequencies; and

wherein, when the first signal is operating at low volume levels, the method provides more low and high frequency components of the first signal than when the first signal is operating a high volume levels;

such that the method provides a transfer function which approximates an inverse of loudness contours for human hearing.

65. (Original) The method of Claim 64, wherein said step of modifying the frequency response comprises the step of increasing the gain of said frequency response in response to an increase in the amplitude level of voice sounds within said signal source.

66. (Original) The method of Claim 64, wherein said signal source is part of a composite multi-channel audio signal and said signal source contains voice sounds mixed with noise.

67. (Previously Presented) A method of emphasizing human speech sounds contained within a signal source to produce an output signal comprises the following steps:

bandpass filtering said signal source to produce a filtered signal wherein said filtered signal includes speech frequencies and attenuates frequencies below and above speech frequencies;

analyzing at least a portion of said filtered signal to produce a control signal wherein said control signal represents a slope of an amplitude envelope of said filtered signal;

amplifying said filtered signal during a first amplification period to provide an enhancement signal wherein the level of amplification of said filtered signal is increased when the slope is positive;

amplifying said filtered signal during a second amplification period to provide an enhancement signal wherein the level of amplification of said filtered signal is decreased when the slope is negative; and

combining said enhancement signal with said signal source to produce an output signal;

wherein said method provides a transfer function that emphasizes middle speech frequencies over low and high frequencies at high volume levels and is flatter at low volume levels, such that said transfer function approximates an inverse of loudness contours for human hearing of tones in a sound field.

68. (Canceled)

69. (Canceled)

70. (Previously Presented) The method of Claim 67, wherein said second amplification period is a function of a predetermined decay time constant.

71. (Original) The method of Claim 67, wherein said signal source is part of a composite signal representing voice and ambient information for presentation to a listener.

72. (Previously Presented) A voice enhancement device for enhancing intelligibility of a voice signal comprising:

a filter configured to receive a voice input signal, the filter configured to attenuate low frequencies below speech frequencies and high frequencies above speech frequencies with respect to speech frequencies to produce a filtered signal;

an envelope detector configured to receive at least a portion of the filtered signal, the envelope detector configured to detect an envelope amplitude of the filtered signal to produce an envelope signal, wherein the envelope signal approximates the envelope amplitude of the filtered signal;

an amplifier configured to receive the filtered signal, the amplifier having a gain control input for controlling a gain of the amplifier, the amplifier configured to amplify the filtered signal according to the gain to produce an amplified signal;

an attack/decay buffer comprising an attack time constant and a decay time constant configured to receive the envelope signal and to produce a gain control signal to control the gain of the amplifier, wherein the attack/decay buffer provides the gain control signal to the gain control input to increase the gain of the amplifier at a rate given by the attack time constant when the envelope signal has a positive slope and to decrease the gain of the amplifier at a rate given by the decay time constant when the envelope signal has a negative slope; and

a combiner configured to add at least a portion of the voice input signal with the amplified signal to produce an enhanced voice signal;

wherein said device is configured to provide a transfer function that approximates an inverse of loudness contours for human hearing of tones in a sound field.

73. (Previously Presented) The device of Claim 72 further comprising a fixed gain amplifier configured to receive the voice input signal and to produce a fixed gain output signal, wherein the fixed gain output signal is combined with the amplified signal.

74. (Previously Presented) The device of Claim 72 wherein the attack time constant is between approximately 1 ms to approximately 40 ms.

75. (Previously Presented) The device of Claim 72 wherein the decay time constant is between approximately 10 ms to approximately 1000 ms.